

(19) World Intellectual Property Organization
International Bureau



(43) International Publication Date
30 November 2000 (30.11.2000)

PCT

(10) International Publication Number
WO 00/72624 A1

(51) International Patent Classification⁷: **H04Q 11/04,**
H04L 12/56

(21) International Application Number: PCT/EP99/03516

(22) International Filing Date: 21 May 1999 (21.05.1999)

(25) Filing Language: English

(26) Publication Language: English

(71) Applicant (for all designated States except US): **NOKIA NETWORKS OY** [FI/FI]; Keilalahdentie 4, FIN-02150 Espoo (FI).

(72) Inventor; and

(75) Inventor/Applicant (for US only): **KOISTINEN, Tommi** [FI/FI]; Kyyhkysmäki 22 B 19, FIN-02600 Espoo (FI).

(74) Agents: **PELLMANN, Hans-Bernd** et al.; Tiedtke-Bühling-Kinne et al., Bavariaring 4, 80336 München (DE).

(81) Designated States (*national*): AE, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, US, UZ, VN, YU, ZA, ZW.

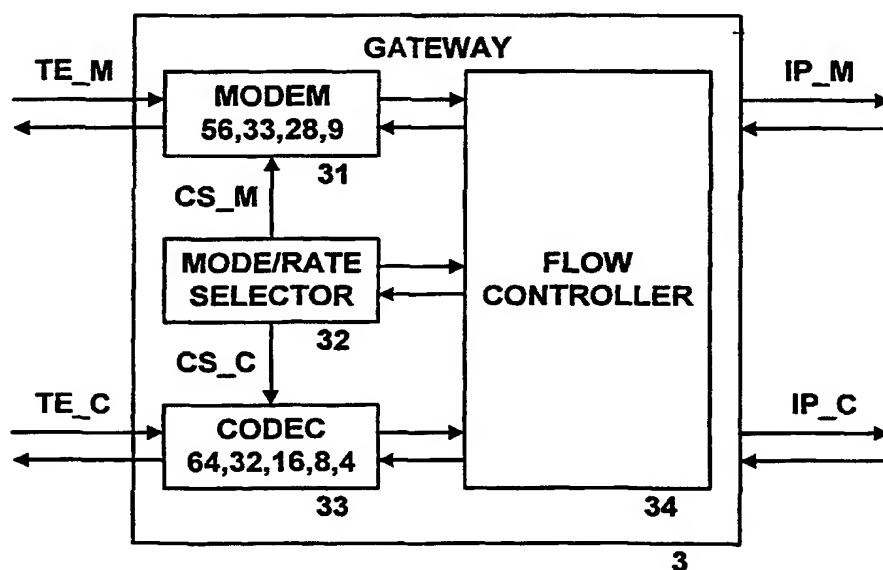
(84) Designated States (*regional*): ARIPO patent (GH, GM, KE, LS, MW, SD, SL, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published:

— With international search report.

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

(54) Title: ADAPTIVE RATE MATCHING FOR DATA OR SPEECH



(57) Abstract: The present invention discloses an interface establishing device (3) for transmitting data to and receiving data from a network, comprising transceiver means (31, 33) being operable with variable transfer rates, a detecting means (34) for detecting the load upon said network (4), and a control means (32) for adjusting the transfer rate of said transceiver means (31, 33) in response to the detected load. By this measure it is possible to adapt the transfer rate of a modem or a codec in response to the load or congestion of a network.



WO 00/72624 A1

ADAPTIVE RATE MATCHING FOR DATA OR SPEECHFIELD OF THE INVENTION

5 The present invention relates to an interface
establishing means and a method for transmitting data to
and receiving data from a network. In particular, the
present invention relates to a gateway between two
different networks and a method for operating such a
10 gateway.

BACKGROUND OF THE INVENTION

In recent years, the Voice over IP (VoIP) technology was
15 developed in which a phone call is sent via an IP-based
network (IP network, Internet Protocol network) such as
the Internet, for example. By sending the signal via such
a network instead of a conventional long distance
carrier, it is possible to reduce the costs involved for
20 such a call.

A general architecture according to the VoIP technology
is shown in Fig. 1. For the purpose of the following
description, the left side of the IP network 4 in Fig. 1
25 is referred to as the near-end side, while the right side
is referred to as the far-end side.

A first communication device 1 such as a mobile phone or
a fixed phone is connected to a first network control
30 device 2 for controlling a first network (near-end
network) to which the mobile phone 1 is connected. The
first network control device 2 is, for example, a mobile
services switching center (MSC). A speech signal is sent
at a bit rate of, e.g., 64 kbps from the first network

- 2 -

control device 2 to a first gateway 3 which connects the near-end network with the IP network 4. The speech signal can be a 64 kbps PCM channel, for example.

5 In order to achieve capacity saving on the IP link, the speech is compressed in the gateway. This compression is performed by a codec (coder-decoder, transcoder, code converter) arranged in the first gateway 3. A typical compression ratio for speech is, for example, 8:1. Since
10 the function of the codec itself is not important to the present invention, a detailed description thereof is omitted here.

The speech signal is compressed, for example, to a bit
15 rate of 8 kbps. The compressed speech signal is sent via the IP network 4 to a second gateway 5. This second gateway also comprises a codec (coder-decoder). However, this codec decompresses the compressed signal received from the IP network 4 to restore the original rate (i.e.,
20 in the above example, 64 kbps). The decompressed speech signal is sent to a second network control device 6 for controlling a second network (far-end network) to which a phone 7 as a second communication device is connected. The second network control device 6 can be a mobile
25 services switching center (MSC) in case the phone 7 is a mobile phone or a fixed services switching center (FSC) in case the phone 7 is a fixed phone. The second network control device 6 sends the signal to the destination phone 7.

30

As described above, the speech signal is compressed and decompressed. In case of a speech signal, this can be effected by using a codec, as described above. The compression serves to save capacity in the IP network.

- 3 -

Furthermore, by compressing the signal, the transmission is not so sensitive to dropped and/or delayed packets as in the case of a non-compressed transmission.

5 Fax and dial-up modems use the same 64 kbps PCM signal as the speech signal does. If such signals (in the following referred to as modem signals) would be processed in the same way as the speech signal (i.e., transmitted via the codec), the modem connections could be blocked
10 completely. For this reason, the gateways 3 and 5 also comprise modems to handle such signals.

In the above situation, high load and even congestion in the IP network is likely to happen, since, for example,
15 the IP network capacity is not overdimensioned in great extent. This will be in particular a problem in case of a further application of the IP telephony in general.

In this situation, any delay caused by the congestion
20 should be minimised. Thus, there is no time for retransmissions of lost packets. Therefore, the UDP (User Datagram Protocol) is commonly used instead of TCP (Transmission Control Protocol). UDP is a rather simple protocol and has a minimum protocol handling. According
25 to this protocol, everything received from the application is sent via the network without any complicated checks. Furthermore, no check is performed whether all data packets have been received by the destination. Thus, this protocol provides a fast, but not
30 very safe transmission.

On the other hand, TCP includes a flow control mechanism. Therefore, this protocol is safer than UDP but requires more protocol handling and more time. This results in a

- 4 -

higher data amount required for the handling of the protocol.

5 In case of an overload or a congestion, UDP is not capable to detect whether any failure in the transmission have occurred. Moreover, in case of a congestion the situation in the IP network is worsen by UDP since the data packets are transferred via the network with a constant rate.

10

In order to make the transmission safer when using UDP, the receiving end can send back in its payloads the information received, such that it can be checked whether the data have been received safely. Alternatively, the
15 UDP could be provided with an acknowledge mechanism like RTCP (Real Time Control Protocol) messages. However, these possibilities both lead to a higher amount of data to be sent via the network, which worsens the congestion situation.

20

Thus, by using the conventional techniques, in case of an overload and congestion of the network, the transmission quality is decreased since packets are delayed or even get lost.

25

SUMMARY OF THE INVENTION

Thus, the object underlying this invention resides in
30 removing the above drawbacks and to enable a sufficient transmission quality even in case of congestion of a network.

- 5 -

This object is solved by an interface establishing device for transmitting data to and receiving data from a network, comprising transceiver means being operable with variable transfer rates, a detecting means for detecting
5 the load upon said network, and a control means for adjusting the transfer rate of said transceiver means in response to the detected load.

Alternatively, the above object is achieved by a method
10 for transmitting data to and receiving data from a network, comprising the steps of detecting the load on said network, and adjusting a transfer rate of a transceiver means in response to said detected load.

15 Thus, it is possible to adapt the transfer rate of a modem or a codec in response to the load or congestion of a network.

That is, in the interface establishing device (gateway)
20 and method for transmitting data to and receiving data from a network according to the present invention, the transfer rate (data rate) can be adapted to the present load on the network. That is, in case a congestion occurs, the transfer rate can be set on a lower value
25 such that data packets can be safely transmitted via the network.

Thus, the transmission quality can be maintained on a sufficient level, since no packet delay or even losses
30 can occur. Only the bandwidth of the speech signal is slightly reduced due to the decreased transfer rate. That is, the speech quality might be reduced slightly, but the end-to-end link stays at least available.

- 6 -

Furthermore, by using the device and the method according to the present invention, it is possible for the IP network to recover faster from a congestion. This is
5 because the transfer rate, i.e., the data amount transmitted per time unit is reduced, such that the load on the network is decreased.

Further advantageous developments of the present
10 invention are stated in the enclosed dependent claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be more readily understood
15 with reference to the accompanying drawings in which:

Fig. 1 shows the basic structure of the VoIP technique;

Fig. 2 shows a gateway according to an embodiment of the
20 present invention; and

Fig. 3 shows a process carried out in the gateway according to the embodiment of the invention.

25

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

The idea of the present invention is to use the congestion indication (or load indication), which is
30 available from the flow control information (e.g., for example from RCTP reports) to control the modem and/or codec transfer rate adaptively. That is, the transfer rate is controlled in such a manner that it is reduced in case a congestion is present and packets get lost and

- 7 -

that it is increased in case no congestion is present and all packets are safely received.

In the following, an embodiment of the invention is
5 described with reference to Figs. 2 and 3.

In Fig. 2, a gateway 3 according to the present embodiment is shown which can be used in the basic VoIP architecture illustrated in Fig. 1. As shown, the gateway
10 3 comprises a modem 31, a codec 33 and a flow controller 34.

The modem 31 serves to compress and decompress fax and/or modem signals TE_M which are transferred to the side of a
15 user terminal. The modem 31 is capable of transmitting with a plurality of different predetermined transfer rates (data rates). For example, the modem could provide transfer rates of 56 kbps, 33 kbps, 28 kbps and 9 kbps. The different rates can be selected by a modem control
20 signal CS_M. The output signal (IP_M) is transferred to the IP network 4 via a flow controller 34.

The codec 33 serves to compress and decompress speech signals TE_D which are transferred to the side of a user
25 terminal (in the configuration of Fig. 1, the phone 1). As the modem 31, the codec 33 is capable of transmitting with a plurality of different predetermined transfer rates (data rates). For example, the codec could provide transfer rates of 64 kbps, 32 kbps, 16 kbps, 8 kbps and 4
30 kbps. The different transfer rates can be selected by a codec control signal CS_C. The output signal (IP_C) is transferred to the IP network 4 via the flow controller 34.

- 8 -

The flow controller 34 serves basically to control the data stream sent to and received from the IP network 4. According to the present embodiment, the flow controller 5 34 also serves to detect the load on the network. The detection can be effected, for example, by using RTCP reports. For example, the (proprietary) RTP/TCP payloads can be used to transfer the number of transmitted/received packets between the gateways 3 and 10 5.

Furthermore, the load can be detected by monitoring Frame Relay's Forward/Backward Explicit Congestion Notification (FECN/BECN) bits, ATM (Asynchronous transfer mode) 15 reports etc.

Moreover, a test packet, for example, an IP PING packet can be sent via the IP network 4 to a predetermined destination, for example to the gateway 5, and then 20 received back from this destination. The occurred delay (round-trip delay) can then be analysed. By such an analysis, a delay can be measured. If this delay suddenly increases from an initially measured level, this indicates a congestion.

25 The flow controller 34 transmits corresponding detection signals to a mode/rate selector 32. According to this detection result, the mode/rate selector 32 sets (adjusts) the transfer rate of the modem 31 and the codec 30 33. For example, the mode/rate selector 32 sets the rate for the codec 33 according to the detected load on the network on 64 kbps PCM, GSM Full Rate (16 kbps) or GSM Half Rate (8kbps). On the other hand, in case of a modem

- 9 -

call this information can be used to adjust the maximum transfer rate of the modem 31.

Furthermore, this information can also serve to adjust
5 the maximum transfer rate between a users modem (which can be arranged in the phone 1 according to Fig. 1, for example) and the modem 31 in the gateway 3 in range of 33,6 kbps, 28,8 kbps, 14,4 kbps and 9,6 kbps by commanding the modem 31. Hence, the amount of data coming
10 from the user towards the IP network 4 can be controlled according to this embodiment.

Fig. 3 shows a flow chart in which a process according to the present embodiment is illustrated.

15

In step S1, the load on the IP network at present is detected. This information is used in step S2, in which the modem transfer rate for the modem 31 and the codec transfer rate for the codec 33 are selected. In step S3,
20 the modem transfer rate determined in this manner is set in the modem 31. Furthermore, in step S4 the determined codec transfer rate is set in the codec 33.

Thus, the transfer rate of the modem 31 and/or the codec
25 33 (which are examples for a transceiver means) can be adapted to the load and the congestion on the IP network.

In the above described embodiment, the modem and the codec have been described as comprising a plurality of
30 different, predetermined transfer rates. However, preferably the transfer rate can be freely (i.e., continuously) adjusted. The more modes (rates) in the modem/codec are, the smoother the transfer rates can be adapted to the load generated in the IP network. Thus,

- 10 -

preferably, a variable bit rate speech codec like an Adaptive Multi Rate (AMR) codec could be used for speech.

In the following, a second embodiment is described, which
5 is a modification of the first embodiment. According to the first embodiment, fixed predetermined transfer rates are set in response to the detected load on the network for both the codec 33 and the modem 31 in the same way. However, it is possible that a lot of non-speech data
10 like fax signals are transmitted via the modem. In this case, a high transmission quality in terms of speed is not as important as in speech signals, since a delay of data packets relating to a fax transmission only
lengthens the time of transmission. In contrast thereto,
15 delay of data packets relating to a speech transmission affect the speech quality greatly.

Thus, according to this embodiment, the transfer via the modem and via the codec are provided with different
20 priorities. That is, in case of an overload or congestion of the IP network, the codec 31 gets a higher transfer rate since the codec mainly transfers speech signals. On the other hand, the modem 33 gets a lower transfer rate since the modem transfers also non-speech signals.

25 Moreover, as a further modification of the above described embodiments, it is also possible to simplify the detection performed by the flow controller 34. Namely, it can be assumed that the load on the IP network
30 does not change abruptly. Thus, it can be sufficient to perform the detection only once in a predetermined period, for example, in every five minutes. For this, a timer can be inserted in the flow controller 34 which outputs an interrupt at the desired time point. In

- 11 -

response to this interrupt, the flow controller 34
performs the process as described with respect to Fig. 3.

Hence, the flow controller 34 does not always have to
5 perform the detection and can be used for other
operations.

The above description and accompanying drawings only
illustrate the present invention by way of example. Thus,
10 the embodiments of the invention may vary within the
scope of the attached claims.

- 12 -

Claims:

1. An interface establishing device for transmitting data to and receiving data from a network (4), comprising
5 transceiver means (31, 33) being operable with variable transfer rates,
a detecting means (34) for detecting the load upon said network (4), and
a control means (32) for adjusting the transfer rate
10 of said transceiver means (31, 33) in response to the detected load.
2. The interface establishing device according to claim 1, wherein in said transceiver means (31, 33) comprise a
15 plurality of predetermined transfer rates and said control means (32) is adapted to select one of said predetermined transfer rates in response to said detected load.
- 20 3. The interface establishing device according to claim 1, wherein said transceiver means comprises a plurality of transceiver means (31, 33).
4. The interface establishing device according to claim
25 3, wherein said control means (32) is adapted to adjust for each of said transceiver means (31, 33) a different transfer rate.
5. The interface establishing device according to claim
30 3, wherein said control means (32) is adapted to provide each of said plurality of transceiver means (31, 33) with different priorities and to adjust a transfer rate of a transceiver means (33) with a higher priority on a higher

- 13 -

value than the transfer rate of a transceiver means (31) with a lower priority

6. The interface establishing device according to one
5 of the previous claims, wherein said transceiver means comprises a modem (31) for modulating and demodulating of non-speech data (TE_M, IP_M).

7. The interface establishing device according to one
10 of the previous claims, wherein said transceiver means comprises a codec (33) for encoding and decoding of speech data (TE_C, IP_C).

8. The interface establishing device according to claim
15 5, wherein said transceiver means comprises a modem (31) for modulating and demodulating of non-speech data (TE_M, IP_M) and a codec (33) for encoding and decoding of speech data (TE_C, IP_C), wherein said control means (32) adapted to provide said codec (33) with a higher priority
20 and to adjust a higher transfer rate for the codec (33) than for the modem (31).

9. The interface establishing device according to one
25 of the previous claims, wherein said control means (32) is adapted to send a test packet to a predetermined destination over said network (4), receive said test packet back from said predetermined destination and analyse the delay occurred in order to determine the load on said network.

30
10. A method for transmitting data to and receiving data from a network (4), comprising the steps of
detecting (S1) the load on said network (4), and

- 14 -

adjusting (**S2, S3, S4**) a transfer rate of a transceiver means (**31, 33**) in response to said detected load.

5 11. The method according to claim 10, wherein in said transceiver means (**31, 33**) comprise a plurality of predetermined transfer rates and in said adjusting step (**S2, S3, S4**) one of said predetermined transfer rates is selected in response to said detected load.

10

12. The method according to claim 10 or 11; wherein said transceiver means comprises a plurality of transceiver means (**31, 33**), and in said adjusting step (**S2, S3, S4**) different transfer rates are set for each of said

15 transceiver means (**31, 33**).

13. The method according to claim 12, further comprising the steps of

20 providing different priorities for each of said plurality of transceiver means (**31, 33**) and

adjusting a transfer rate of a transceiver means (**33**) with a higher priority on a higher value than the transfer rate of a transceiver means (**31**) with a lower priority.

25

14. The method according to one of the claims 10 to 13, wherein said transceiver means comprises a modem (**31**) for modulating and demodulating of non-speech data (**TE_M, IP_M**).

30

15. The method according to one of the claims 10 to 14, wherein said transceiver means comprises a codec (**33**) for encoding and decoding of speech data (**TE_C, IP_C**).

- 15 -

16. The method according to one of the claims 10 to 13,
wherein said transceiver means comprises a modem (**31**) for
modulating and demodulating of non-speech data (**TE_M**,
5 **IP_M**) and a codec (**33**) for encoding and decoding of
speech data (**TE_C**, **IP_C**), further comprising the steps of
providing said codec (**33**) with a higher priority and
adjusting a transfer rate of the codec (**33**) on a
higher value than the transfer rate of the modem (**31**).

10

17. The method according to one of the claims 10 to 16,
further comprising the steps of
sending a test packet to a predetermined destination
over said network (**4**);
15 receiving said test packet back from said
predetermined destination; and
analysing the delay occurred in order to determine
the load on said network.

1/2

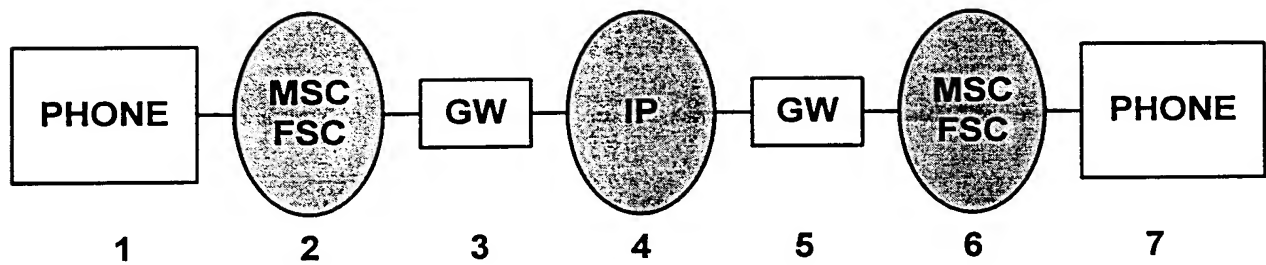


FIG. 1

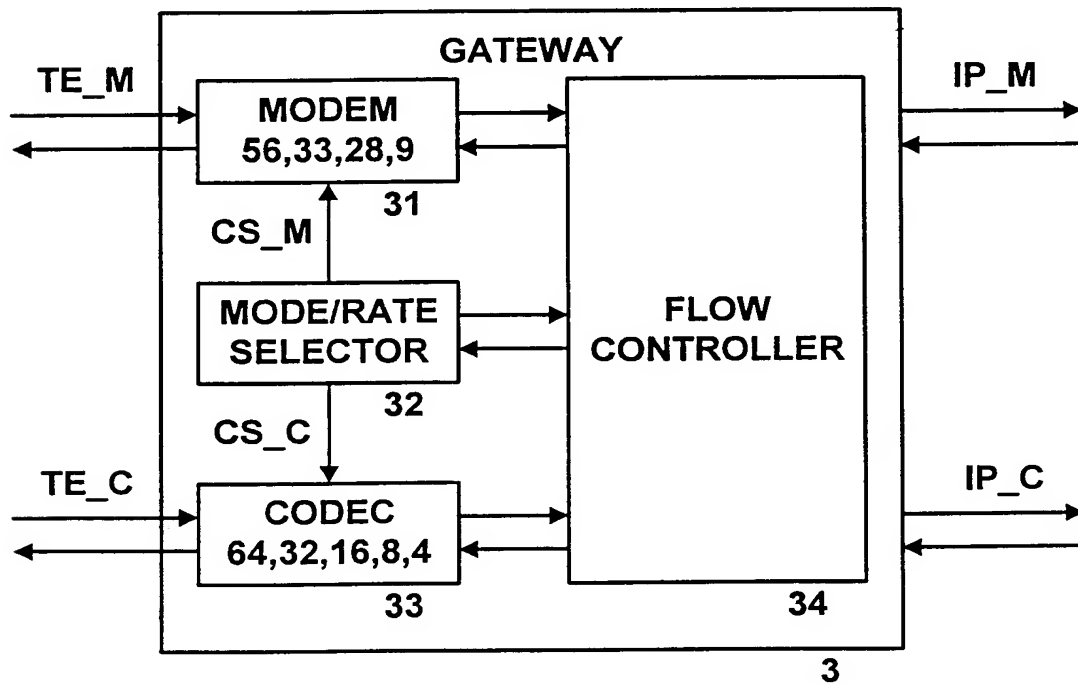
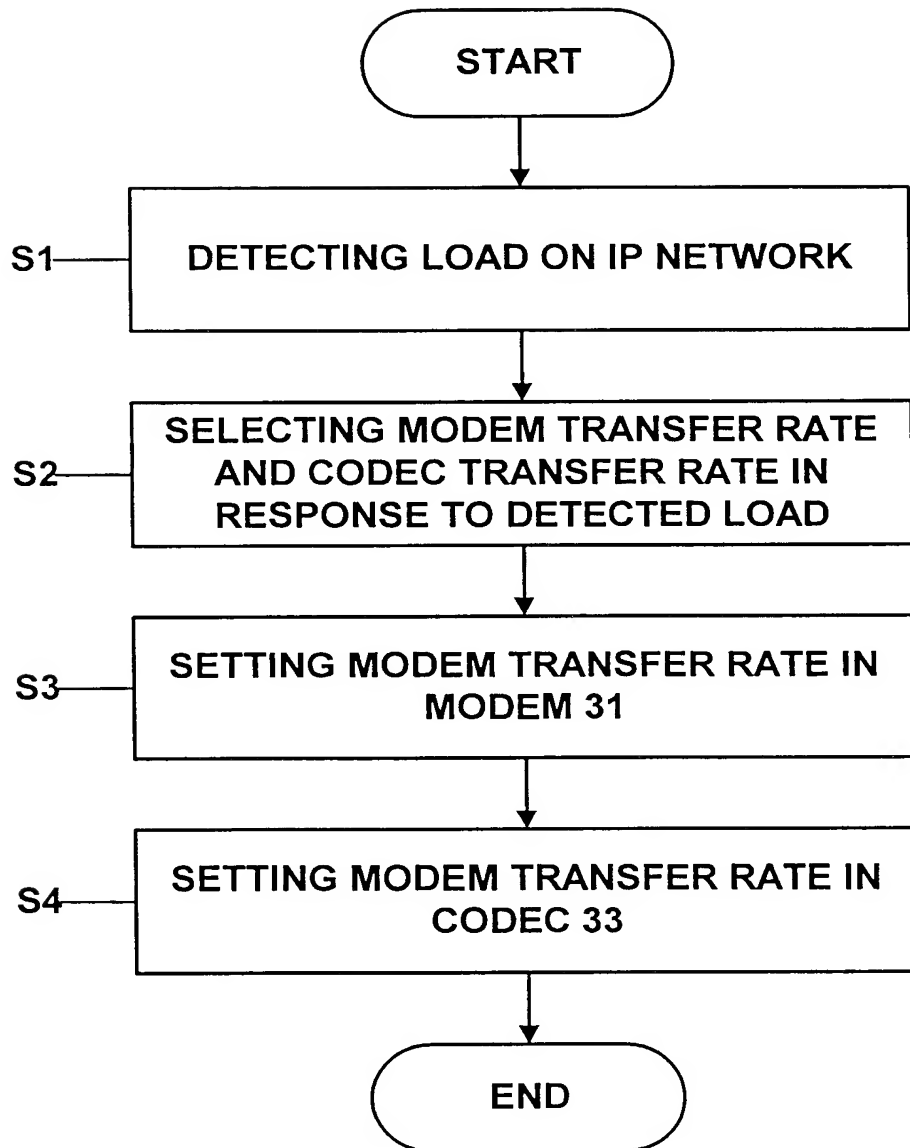


FIG. 2

**FIG. 3**

INTERNATIONAL SEARCH REPORT

International Application No
PCT/EP 99/03516

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04Q11/04 H04L12/56

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0 782 302 A (LUCENT TECHNOLOGIES INC) 2 July 1997 (1997-07-02) column 1, line 30-58 column 2, line 15 -column 3, line 5 column 5, line 44 -column 6, line 10 column 7, line 14-42 claims 1-3	1-4, 10-12
Y	---	5,6,9, 13,14,17
	-/--	

☒ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search

21 October 1999

Date of mailing of the international search report

05/11/1999

Name and mailing address of the ISA
European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
Fax: (+31-70) 340-3016

Authorized officer

Kalabic, F

INTERNATIONAL SEARCH REPORT

International Application No
PCT/EP 99/03516

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category ²	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	NANYING YIN ET AL: "A DYNAMIC RATE CONTROL MECHANISM FOR INTEGRATED NETWORKS" , NETWORKING IN THE NINETIES, BAL HARBOUR, APR. 7 - 11, 1991, VOL. VOL. 2, NR. CONF. 10, PAGE(S) 543 - 552 , INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS XP000223378ISBN: 0-87942-694-2 abstract page 545, left-hand column, paragraph 1 page 546, left-hand column, paragraph 4 page 548, left-hand column, paragraph 3 -right-hand column, paragraph 1 page 549, left-hand column, paragraph 3 page 549, right-hand column, line 7 - line 16 ---	1-3,7,15
X	EP 0 790 725 A (FUJITSU LTD) 20 August 1997 (1997-08-20) column 1, line 18-21 column 3, line 23-28 ---	1,7,15
A	---	8,16
Y	EP 0 706 297 A (IBM) 10 April 1996 (1996-04-10) abstract page 4, line 55-59 page 5, line 1-3 ---	5,13
A	---	8,16
Y	HAAS Z: "ADAPTIVE ADMISSION CONGESTION CONTROL" , COMPUTER COMMUNICATIONS REVIEW, VOL. 21, NR. 5, PAGE(S) 58 - 76 XP000240808 ISSN: 0146-4833 page 62, line 6 - line 31 ---	9,17
Y	US 5 805 591 A (GHAIBEH GIHAD ET AL) 8 September 1998 (1998-09-08) abstract column 2, line 30 - line 45 ---	6,14
A	-----	8,16

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/EP 99/03516

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0782302 A	02-07-1997	US 5701292 A JP 9181744 A	23-12-1997 11-07-1997
EP 0790725 A	20-08-1997	JP 2865782 B JP 3267846 A CA 2038436 A,C DE 69131365 D EP 0446956 A US 5544170 A	08-03-1999 28-11-1991 17-09-1991 29-07-1999 18-09-1991 06-08-1996
EP 0706297 A	10-04-1996	US 5790522 A	04-08-1998
US 5805591 A	08-09-1998	AU 1965897 A EP 0893015 A WO 9732411 A	16-09-1997 27-01-1999 04-09-1997